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EXAMINER	
PARK, JEONG S	

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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No. 10/761,012	Applicant(s) HOLLATZ, MIKE	
	Examiner Jeong S. Park	Art Unit 2454	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 30 July 2010.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-27 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-27 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

1. This communication is in response to Application No. 10/761,012 filed on 1/20/2004. The amendment presented on 7/30/2010, which amends claims 1, 5, 10, and 19, is hereby acknowledged. Claims 1-27 have been examined.

Claim Rejections - 35 USC § 103

2. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

3. Claims 1-5, 10-14 and 19-23 are rejected under 35 U.S.C. 103(a) as being unpatentable over Holden et al. (hereinafter Holden)(U.S. Patent No. 7,233,980) in view of Foti (U.S. Pub. No. 2002/0194378), and further in view of Baruch (U.S. Patent No. 6,570,980).

Regarding claims 1, 10 and 19, Holden teaches as follows:

A method of routing a SIP call within an automatic contact distributor system wherein an initial SIP message of the SIP call from a caller is forwarded to the automatic contact distributor system via a first server (interpreted as a SIP server, ACD 62 includes a SIP server, see, e.g., col. 8, lines 50 and figure 4)(The method and system of the present disclosure can be used for ACD services developed for IP

telephony using SIP as the signaling protocol, see, e.g., col. 2, line 61 to col. 3, line 8), such method comprising the steps of:

selecting an agent of a plurality of agents of the automatic contact distributor system to handle the SIP call forwarded to the automatic contact distributor from the first server (When the PBX 102 indicates to the ACD 108 that an agent 112 is available to answer a call, see, e.g., col. 9, lines 4-10); and

setting up a the SIP call connection between an the selected agent of the automatic call distributor system and the caller so as to route any SIP messages between the agent and the caller through the first server (the caller can get transferred to a live agent 50 by the ACD system, as indicated by reference numeral 49 in the flow diagram of FIG. 6., see, e.g., col. 9, lines 45-64).

Holden does not teach of the second server located between the agent and the caller for modifying any source addresses of the SIP messages sent from the agent to the caller and received by the second server from the agent by substituting an address of the second server in SIP messages sent from the agent to the caller and forwarding the modified SIP messages to the caller thereby protecting anonymity of the agent from the caller by concealing URLs and any other identification information of the agent and re-addressing SIP messages received from the caller at the second server and forwarding the re-addressed SIP messages to the agent.

Foti teaches as follows:

A system and method of hiding the source Internet Protocol (IP) address of an originating and/or terminating terminal during media flow by routing IP packets through

an enhanced Media Resource Function (MRF equivalent to applicant's second server or buffer server) that removes the source address and substitutes an alias address (see, e.g., abstract); and

alternatively, the terminating IP terminal may wish to hide its IP address from the originating IP terminal. In this case, an address translation function in the home network of the terminating IP terminal replaces the source address of the terminating IP terminal with the IP address of the address translation function (see, e.g., page 1, paragraph [0008]);

the present invention is described herein primarily in terms of the Session Initiation Protocol (SIP) developed by the Internet Engineering Task Force (IETF), but is equally applicable to the International Telecommunications Union (ITU) H.323 protocol, or other packet-switched control protocols. In a typical IP network, PC clients or IP telephony terminals (fixed or mobile) are identified and addressed by an e-mail address (proxy/alias), or an IP address or both. The present invention makes a substitution for this identifying address, regardless of the specific protocol (see, e.g., page 4, paragraph [0030]);

modifying any source addresses of the SIP messages sent from the agent (terminal B, 13 in figure 4) to the caller (terminal A, 10 in figure 4)(MRF-M modifies the terminal B's address to $IPTF_B$ and sends to the terminal A, see, e.g., page 6, paragraph [0049] and step 107 and 109 in figure 4) and received by the second server (interpreted as MRF-M) from the agent by substituting an address of the second server ($IPTF_B$) in SIP messages sent from the agent to the caller (the MRF-M substitutes $IPTF_B$ as the

source address, see, e.g., page 6, paragraph [0049]) and forwarding the modified SIP messages to the caller thereby protecting anonymity of the agent from the caller by concealing URLs and any other identification information of the agent (see, e.g., step 109 in figure 4 and page 6, paragraph [0048] and [0049]); and

re-addressing SIP messages received from the caller at the second server and forwarding the re-addressed SIP messages to the agent (MRF-M re-addresses destination address $IPTF_B$ with address B, see, e.g., step 104 in figure 4 and page 6, paragraph [0048]).

It would have been obvious for one of ordinary skill in the art at the time of the invention to combine Holden with Foti to include an enhanced Media Resource Function that removes the source address and substitutes an alias address as taught by Foti in order to efficiently hide the source address of an originating or terminating terminal.

Holden in view of Foti does not explicitly teach of selecting an agent of a plurality of agents of the automatic contact distribution system.

Baruch teaches as follows:

A method of distributing telephone calls to provide users of extensions and/or terminals of a telecommunications network with access to a service provided by an agent includes the steps of determining agent profiles, determining a call profile, comparing the call profile to the agent profiles to constitute an ordered list of agents qualified to process a call, and distributing a call to agents from the list of agents (see, e.g., abstract); and

in this second step each call entering a waiting room and which has previously been associated with a call profile initiates an agent selection operation (see, e.g., col. 7, lines 25-49).

It would have been obvious for one of ordinary skill in the art at the time of the invention to combine Holden in view of Foti with Baruch to include a method of selecting an agent in distributing telephone calls as taught by Baruch in order to efficiently distribute a call to agents from the list of agents (see, e.g., Baruch, abstract).

Regarding claims 2, 11 and 20, Holden teaches as follows:

Receiving a SIP INVITE (see, e.g., col. 8, lines 15-20) from the caller by the automatic contact distributor system requesting a communication session with an agent of the automatic contact distributor system (In one embodiment, a SIP based client is utilized for establishing the call with the ACD server and the ACD server being within a PSTN, see, e.g., col. 11, lines 1-11).

Regarding claims 3, 12 and 21, Holden teaches as follows:

Determining a call type from the SIP INVITE (SIP invitations, used to create sessions, carry session descriptions, which allow participants to agree on a set of compatible media types, see, e.g., col. 5, lines 59-66).

Regarding claims 4, 13 and 22, Holden teaches as follows:

When the PBX 102 indicates to the ACD 108 that an agent 112 is available to answer a call (see, e.g., col. 9, lines 4-10).

Because SIP invitations carry session descriptions, which allow participants to agree on a set of compatible media types (see, e.g., col. 5, lines 59-66), it would have

been obvious for one of ordinary skill in the art at the time of the invention to modify Holden to select an agent based on media types indicated with SIP invitations.

Regarding claims 5 and 14, Holden teaches of SIP message exchange between the caller and the agent as presented above except for using a buffer server substituting a source URL of the buffer server in SIP messages sent from the agent to the caller.

Foti teaches as follows:

Forwarding the SIP INVITE to the buffer server along with an identifier of the selected agent and the buffer server substituting a source URL of the buffer server in SIP messages sent from the agent to the caller (MRF-M (interpreted as the second server and buffer server) modifying the terminal B's address to $IPTF_B$ and sends to the terminal A, see, e.g., page 6, paragraph [0049] and step 107 and 109 in figure 4).

Therefore, Holden in view of Foti teaches that the second server is a buffer server and the step of setting up the call further comprises forwarding the SIP INVITE to the buffer server along with an identifier of the selected agent.

Regarding claim 23, Foti teaches as follows:

A routing table for re-addressing the SIP messages that are transferred between the agent and the client (address translation table, see, e.g., page 7, paragraph [0047] and table 2).

Therefore, it is rejected for similar reason as presented above in claim 19.

4. Claims 6, 7, 15, 16, 24 and 25 are rejected under 35 U.S.C. 103(a) as being unpatentable over Holden et al. (hereinafter Holden)(U.S. Patent No. 7,233,980) in view

of Foti (U.S. Pub. No. 2002/0194378) and Baruch (U.S. Patent No. 6,570,980), and further in view of Wengrovitz (U.S. Pub. No. 2002/0141404).

Regarding claims 6 and 15, Wengrovitz teaches as follows:

Entering the SIP INVITE into a routing table within the buffer server along with the identifier (SIP URL) of the selected agent (location server 25 in figure 1A retrieves the SIP URL associated with the called end-point to resolve the URL to a more precise address, see, e.g., page 1, paragraph [0006]), therefore the location server inherently includes a table to map between the SIP URL and the more precise address.

It would have been obvious for one of ordinary skill in the art at the time of the invention to combine Holden in view of Foti and Baruch with Wengrovitz to include the conventional SIP sessions as taught by Wengrovitz in order to utilize a SIP messaging structure within a call center's architecture for interacting with a caller (see, Holden, abstract).

Regarding claims 7, 16 and 25, Wengrovitz teaches as follows:

Appending the identifier (SIP URL) to a universal resource identifier (specific IP address) of the buffer server (location server 100 in figure 5) within the SIP INVITE (location server deduces the address using information in the location server and database for ascertaining a most appropriate IP address, see, e.g., page 4, paragraph [0041]).

Therefore, they are rejected for similar reason as presented above in claims 6 and 15.

Regarding claim 24, Wengrovitz teaches as follows:

the proxy server further comprises an Internet connection (Internet 54 in figure 2, see, e.g., page 3, paragraph [0031]) that allows the proxy server to forward the SIP INVITE to the buffer server along with an identifier of the selected agent (the "To" filed of the SIP INVITE message header includes a generic SIP URL associated with a called end-point, see, e.g., page 1, paragraph [0005]).

Therefore, they are rejected for similar reason as presented above in claims 6 and 15.

5. Claims 8, 17 and 26 are rejected under 35 U.S.C. 103(a) as being unpatentable over Holden et al. (hereinafter Holden)(U.S. Patent No. 7,233,980) in view of Foti (U.S. Pub. No. 2002/0194378) and Baruch (U.S. Patent No. 6,570,980), and further in view of Strathmeyer et al. (hereinafter Strathmeyer)(U.S. Patent Pub. No. 2004/0120502 A1).

Regarding claims 8, 17 and 26, Holden in view of Foti and Baruch teaches all the limitations of claim except for explicitly showing conversion from SIP protocol to instant message protocol.

Strathmeyer teaches as follows:

gateway (120 in figure 1) may receive a call setup request signal from a PSTN network and then generate and send a corresponding SIP INVITE message, which may request the setup of a corresponding packet telephony call (see, e.g., page 4, paragraph [0038]); and

gateway may provide protocol conversion or protocol interworking between any types of protocols (see, e.g., page 4, paragraph [0039]).

It would have been obvious for one of ordinary skill in the art at the time of the invention to combine Holden in view of Foti and Baruch with Strathmeyer to include gateway functionality capable of protocol conversion between SIP protocol and instant message protocol as taught by Strathmeyer in order to widely utilize the automatic call routing method for any different networks environment.

6. Claims 9, 18 and 27 are rejected under 35 U.S.C. 103(a) as being unpatentable over Holden et al. (hereinafter Holden)(U.S. Patent No. 7,233,980) in view of Foti (U.S. Pub. No. 2002/0194378) and Baruch (U.S. Patent No. 6,570,980), and further in view of Borella et al. (hereinafter Borella)(U.S. Patent No. 6,816,912 B1).

Regarding claims 9, 18 and 27, Holden in view of Foti and Baruch teaches all the limitations of claim except for using tunneling protocol for communication between the client and the buffer server.

Borella teaches as follows:

a method and system for tunnel optimized call setup for mobile nodes (see, e.g., col. 2, line 61 to col. 3, line 13 and abstract).

It would have been obvious for one of ordinary skill in the art at the time of the invention to combine Holden in view of Foti and Baruch with Borella to include tunneling method and system for call setup as taught by Borella in order to optimize the call setup process between different networks.

Response to Arguments

7. Applicant's arguments filed 7/30/2010 have been fully considered but they are not persuasive.

A. Summary of Applicant's Arguments

In the remarks, the applicant argues as followings:

1) Regarding amended claims 1, 10, and 19, Holden in view of Foti and Baruch does not teach a call sent to an ACD thru the first server, selecting one of a plurality of agents, setting up a SIP call between the selected agent and the caller after the ACD system receives the initial SIP message, or setting up the SIP call through a second server or a buffer server.

2) Regarding amended claims 1, 10, and 19, Holden in view of Foti and Baruch does not teach a second server modifying the source address of the selected agent's SIP messages by substituting that of the second or buffer server to thereby maintain anonymity of the agent by concealing the URL and any other identifying information of the agent.

3) Regarding amended claims 6 and 15, Holden in view of Foti and Wengrovitz does not teach sending a SIP INVITE to a buffer server, or entering the forwarded message into a routing table in the buffer server.

B. Response to Arguments:

In response to argument 1), Holden teaches as follows:

a call sent to an ACD thru the first server (the first server can be any server between a caller and an ACD. Therefore, PSTN-to-IP gateway 68 in figure 4 or a server used in the Internet 66 in figure 4 can be equivalent to applicant's first server, see, e.g., col. 8, lines 43-54 and figure 4);

selecting one of a plurality of agents (one agent is selected from multiple agents, see, e.g., col. 9, lines 4-10 and 112 in figure 5); and

setting up a SIP call between the selected agent and the caller (ACD services developed for IP telephony using SIP as the signaling protocol, see, e.g., col. 2, line 61 to col. 3, line 8) after the ACD system receives the initial SIP message (route the call to he available agent, see, e.g., col. 9, lines 4-10 and 112 in figure 5).

Foti teaches as follows:

modifying any source addresses of the SIP messages sent from the agent (terminal B, 13 in figure 4) to the caller (terminal A, 10 in figure 4)(MRF-M modifies the terminal B's address to $IPTF_B$ and sends to the terminal A, see, e.g., page 6, paragraph [0049] and step 107 and 109 in figure 4) and received by the second server (interpreted as MRF-M) from the agent by substituting an address of the second server ($IPTF_B$) in SIP messages sent from the agent to the caller (the MRF-M substitutes $IPTF_B$ as the source address, see, e.g., page 6, paragraph [0049]) and forwarding the modified SIP messages to the caller thereby protecting anonymity of the agent from the caller (see, e.g., step 109 in figure 4 and page 6, paragraph [0048] and [0049]); and

re-addressing SIP messages received from the caller at the second server and forwarding the re-addressed SIP messages to the agent (MRF-M re-addresses

destination address $IPTF_B$ with address B, see, e.g., step 104 in figure 4 and page 6, paragraph [0048]).

Therefore, Foti teaches of setting up the SIP call through a second server or a buffer server (interpreted as MRF-M, 21b in figure 2).

In response to argument 2), Foti teaches as follows:

modifying any source addresses of the SIP messages sent from the agent (terminal B, 13 in figure 4) to the caller (terminal A, 10 in figure 4)(MRF-M modifies the terminal B's address to $IPTF_B$ and sends to the terminal A, see, e.g., page 6, paragraph [0049] and step 107 and 109 in figure 4) and received by the second server (interpreted as MRF-M) from the agent by substituting an address of the second server ($IPTF_B$) in SIP messages sent from the agent to the caller (the MRF-M substitutes $IPTF_B$ as the source address, see, e.g., page 6, paragraph [0049]) and forwarding the modified SIP messages to the caller thereby protecting anonymity of the agent from the caller by concealing the URL and any other identifying information of the agent (see, e.g., step 109 in figure 4 and page 6, paragraph [0048] and [0049]).

In response to argument 3), Foti teaches of sending a SIP INVITE to a buffer server (interpreted as MRF-M) as presented above.

Wengrovitz teaches as follows:

Entering the forwarded message into a routing table in the buffer server (the proxy server accepts the INVITE request and in step 32, preferably engages a location

server 25 for routing the call based on the routing information in the SIP message header, see, e.g., page 1, paragraph [0006]). Therefore the location server inherently includes a routing table to map between the SIP URL and the more precise address.

Conclusion

8. **THIS ACTION IS MADE FINAL.** Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of this final action.

9. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Jeong S. Park whose telephone number is (571)270-1597. The examiner can normally be reached on Monday through Friday 9:00 - 5:30 EST.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Joseph E. Avellino can be reached on 571-272-3905. The fax phone

number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

/J. S. P./
Examiner, Art Unit 2454

October 8, 2010

/Joseph E. Avellino/

Supervisory Patent Examiner, Art Unit 2454